REMARKS

Applicants have carefully reviewed this application in light of the Final Office Action mailed August 25, 2004, and the Advisory Action mailed December 1, 2004. Applicants appreciate the Examiner's consideration of the Application and respectfully request favorable action in this case

Claim Rejections -- 35 U.S.C. § 102

The Examiner rejected Claims 2-7, 39, 9-14, 40, 32-34, 37-38, and 40 under 35 U.S.C. §102(e) as being anticipated by U.S. Patent No. 6,463,414 ("*Su*").

The entire disclosure of *Su* is not prior art under § 102(e) because it was filed on April 12, 2000, which is almost four months after Applicants' filing date of December 15, 1999. While *Su* claims priority to a provisional application filed on April 12, 1999, that provisional application does not support the entire disclosure of *Su*. *Su* may qualify as prior art only to the extent the disclosure is supported by Provision Application No. 09/547,832 ("*Su Provisional Application*"), which is attached for the Examiner's consideration.

Independent Claim 38 and Dependent Claims 2-7

Independent Claim 38 recites:

> An apparatus for using a plurality of processors to support a media conference, comprising:
>
> a mixing processor operable to mix input media information associated with two or more first participants to generate output media information for communication to a second participant; and
>
> a first media transformation processor coupled to the mixing processor, the first media transformation processor operable to receive the output media information from the mixing processor, to encode the output media information to generate an output data stream, and to communicate the output data stream to the second participant's end-user device.

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Su does not disclose, teach, or suggest Applicants' claimed invention because, as the Examiner has acknowledged, *Su* does not disclose separate processors as recited in the claims.

Independent Claims 38 requires multiple processors: a "mixing processor" and a "first media transformation processor." The specification provides:

> Media transformation processors 12 and mixing processors 14 represent separate hardware components. The functionality described below may be implemented using separate hardware components or software that executes using the separate hardware components. Thus, media transformation processor 12 and mixing processor 14 do not operate using the same actual physical computing machinery. Media transformation processors 12 and mixing processors 14 may represent separate microprocessors, controllers, digital signal processors (DSPs), or other integrated circuit chips mounted to a circuit board. Alternatively, media transformation processors 12 and mixing processors 14 may represent separate networks of electronic components, such as transistors, diodes, resistors, etc., and their interconnections etched or imprinted on a single chip. In such an embodiment, media transformation processors 12 and mixing processors 14 may use shared resources but generally rely on separate pipelines to perform the majority of their processing. Although media transformation processors 12 and mixing processors 14 represent separate hardware components, the hardware components are not necessarily different in type. In a particular embodiment, media transformation processors 12 and mixing processors 14 are implemented using the same type of digital signal processors.

(p. 9). Thus, a "mixing processor" and a "first media transformation processor" represent separate hardware components.

The Examiner has several times acknowledged that *Su* does not disclose separate processors. In the Office Action mailed February 25, 2004, the Examiner stated:

> Su did no specifically disclose said processors being separate as in claim 35, being DSP as in claim 36.

(p. 5). The Examiner repeated this statement in the Office Action mailed August 25, 2004. (p. 5).

The Examiner was correct in his observation. *Su* expressly states that the invention is described in terms of "functional block components" which may be implemented using "any number of hardware components or software elements":

> The present invention may be described herein in terms of functional block components and various processing steps. It should be appreciated that such functional blocks may be realized by any number of hardware components or software elements configured to perform the specified functions. For example, the present invention may employ various integrated circuit components, e.g., memory elements, digital signal processing elements, logic elements, look-up tables, and the like, which may carry out a variety of functions under the control of one or more microprocessors or other control devices.

(Col. 2, ll. 49-59) (emphasis added). *Su* does not specify that the functions of decoders 230 and 234, mixer 238 and 240, and encoder 232 and 236 are assigned to separate processors. Indeed, *Su* provides that the functional blocks may be implemented in software.

Another passage in *Su* further indicates that Figure 2, on which the Examiner relies to support the rejections, is a "simplified schematic" of functional blocks as opposed to hardware components.

> FIG. 2 is a simplified schematic: there might also be certain additional components advantageously coupled between the packet network and the decoders (and encoders). Specifically, with respect to the decoders, there. will likely be a functional block (not shown) that receives the packets from packet network 201 and removes all unnecessary routing, encryption, and protection information (a "decapsulator"). Conversely, with respect to the encoders, there will likely be a functional block (an "encapsulator") for each encoder that receives speech samples from the mixer and adds certain information regarding routing, encryption, and the like prior to sending the packets out over packet network 201.

(Col. 5, ll. 19-31) (emphasis added).

Furthermore, Figure 2 of *Su* is not prior art because it is not included in the *Su Provisional Application*. Figure 2 of the attached *Su Provisional Application* (which may or may not be prior art) even more clearly portrays the decoding, mixing and re-encoding as functional blocks as opposed to separate hardware components.

For at least these reasons, *Su* does not disclose, teach, or suggest the "mixing processor" and "first media transformation processor" of Claim 38. Accordingly, Applicants respectfully request reconsideration and allowance of independent Claims 38, as well as Claims 2-7 which depend from Claim 38.

<u>Independent Claim 39 and Dependent Claims 9-14</u>

Independent Claim 39 recites:

> A method for using a plurality of processors to support a media conference, comprising:
>
> mixing input media information associated with two or more first participants to generate output media information for communication to a second participant;
>
> communicating the output media information from a mixing processor to a first media transformation processor;
>
> encoding the output media information to generate an output data stream; and
>
> communicating the output data stream from the first media transformation processor to the second participant's end-user device.

Su does not disclose, teach, or suggest Applicants' claimed invention because, as the Examiner acknowledged in the Office Action, *Su* does not disclose separate processors as recited in the claims. Like Claims 38, independent Claim 39 requires multiple processors. Claim 39 recites the steps "mixing input media information associated with two or more first participants to generate output media information for communication to a second participant," "communicating the output media information from a <u>mixing processor</u> to a <u>first media transformation processor</u>," and "encoding the output media information to generate an output data stream." As pointed out above with respect to Claim 38, these separate processors represent separate hardware components. Because *Su* and the *Su Provisional Application* disclose functional blocks as opposed to separate processors, *Su* does not disclose, teach, or suggest the "mixing processor" and "first media transformation processor" of Claim 39. Accordingly, Applicants respectfully request reconsideration and allowance of independent Claims 39, as well as Claims 9-14 which depend from Claim 39.

Independent Claim 40 and Dependent Claims 32-34 and 37

Independent Claim 40 recites:

> A system for using a plurality of processors to support a media conference, comprising:
>
> a plurality of end-user devices coupled to a data network and operable to generate input media information, to encode the input media information to generate input data streams, and to communicate the input data streams using the data network; and
>
> a conferencing device coupled to the data network, the conferencing device comprising two or more processors operable to decode the input data streams to generate the input media information, to mix the input media information to generate output media information, and to encode the output media information to generate output data streams;
>
> wherein the end-user devices are further operable to receive the output data streams and to decode the output data streams to generate output media information

Su does not disclose, teach, or suggest Applicants' claimed invention because, as the Examiner acknowledged in the Office Action, *Su* does not disclose separate processors as recited in the claims. Like Claims 38 and 39, independent Claim 40 requires multiple processors. Claim 40 recites, "the conferencing device comprising two or more processors operable to decode the input data streams to generate the input media information, to mix the input media information to generate output media information, and to encode the output media information to generate output data streams." As pointed out above with respect to Claim 38, these separate processors represent separate hardware components. Because *Su* and the *Su Provisional Application* disclose functional blocks as opposed to separate processors, *Su* does not specify that the functions of decoders 230 and 234, mixer 238 and 240, and encoder 232 and 236 are assigned to separate processors. For at least this reason, *Su* does not disclose, teach, or suggest "the conferencing device comprising two or more processors operable to decode the input data streams to generate the input media information, to mix the input media information to generate output media information, and to encode the output media information to generate output data streams," as recited in Claim 40.

Accordingly, Applicants respectfully request reconsideration and allowance of independent Claims 40, as well as Claims 32-34 and 37 which depend from Claim 40.

Claim Rejections -- 35 U.S.C. § 103

The Examiner rejected Claims 5, 6, and 35-36 under 35 U.S.C. § 103 as being unpatentable over *Su* in view of U.S. Patent 5,841,763 ("*Leondires*").

According to the Examiner, *Leondires* "discloses a conferencing device with separate processors." (p. 5). *Leondires*, however, does not disclose, teach, or suggest using separate processors for mixing and encoding. The portion of the specification cited by the Examiner describes audio encoding digital signal processors (ADPs) and audio encoding digital signal processors (AEPs). The ADPs decode audio information. (Col. 14, ll. 33-43). The AEPs mix and encode audio information: "The AEPs read the decoded audio signals from DSs time slots, mix the decoded audio signals from each of the conferees and encode the results of the mixing according to the particular G-series standard." (Col. 14, ll. 51-54).

In contrast to the AEPs of *Leondires*, Claims 38 and 39 require two separate processors for mixing and encoding. Claim 39 requires: (1) "a mixing processor operable to mix input media information" and (2) "first media transformation processor operable to receive the output media information from the mixing processor, to encode the output media information to generate an output data stream, and to communicate the output data stream to the second participant's end-user device." Similarly, Claim 39 distinguishes between a mixing processor for mixing and a media transformation processor for encoding. Claim 39 requires the following steps: "mixing input media information associated with two or more first participants to generate output media information for communication to a second participant," "communicating the output media information from a mixing processor to a first media transformation processor," and "encoding the output media information to generate an output data stream."

For the reasons discussed above with respect to independent Claims 38, 39, and 40, as well as these additional reasons, *Su* and *Leondires* do not disclose Applicants' claimed invention recited in dependent Claims 5, 6, and 35-56. Accordingly, Applicants respectfully request reconsideration and allowance of dependent Claims 5, 6, and 35-36.

CONCLUSION

Applicants have made an earnest attempt to place this case in condition for allowance. For the foregoing reasons, and for other reasons clearly apparent, Applicants respectfully request full allowance of pending Claims 2-7, 9-14, and 32-40. If the Examiner feels that a telephone conference or an interview would advance prosecution of this Application in any manner, the undersigned attorney for Applicants stands ready to conduct such a conference at the convenience of the Examiner.

Applicants enclose a check for $790.00 to cover the filing of this Request for Continued Examination (RCE). Applicants also enclose a check for $120.00 to cover the cost of filing a one-month extension of time. The Commissioner is hereby authorized to charge any other fees or credit any overpayments to Deposit Account No. 02-0384 of Baker Botts L.L.P.

Respectfully submitted,

BAKER BOTTS L.L.P.
Attorneys for Applicants

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Correspondence Address:

_X_ Customer Number **05073**

Date: January 3, 2005

Please type a plus sign (+) inside this box → +

Jc658 U.S. PTO

Docket Number:	50944.4600

PROVISIONAL APPLICATION FOR PATENT COVER SHEET (Large Entity)

This is a request for filing a PROVISIONAL APPLICATION FOR PATENT under 37 CFR 1.53 (c).

Jc549 U.S. PTO 60/128873 04/12/99

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Adil	Benyassine	
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☐ *Additional inventors are being named on page 2 attached hereto*

TITLE OF THE INVENTION (280 characters max)

METHOD AND APPARATUS FOR CONFERENCE BRIDGE PROCESSING OF SPEECH

CORRESPONDENCE ADDRESS

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ENCLOSED APPLICATION PARTS *(check all that apply)*

☒ Specification — *Number of Pages* 8

☒ Drawing(s) — *Number of Sheets* 4

☐ Other (specify)

METHOD OF PAYMENT OF FILING FEES FOR THIS PROVISIONAL APPLICATION FOR PATENT *(check one)*

☒ A check or money order is enclosed to cover the filing fees

☐ The Commissioner is hereby authorized to charge filing fees or credit any overpayment to Deposit Account Number:

FILING FEE AMOUNT: $150.00

The invention was made by an agency of the United States Government or under a contract with an agency of the United States Government.

☒ No

☐ Yes, the name of the U S Government agency and the Government contract number are: ______

Respectfully submitted,

SIGNATURE M M. Tak — DATE April 12, 1999

TYPED or PRINTED NAME Mark M. Takahashi — REGISTRATION NO. *(if appropriate)* 38,631

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P19LARGE/REV04

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I certify that this provisional patent application cover sheet, provisional patent application and fee is being deposited on April 12, 1999 with the U.S. Postal Service as "Express Mail Post Office to Addressee" service under 37 C.F.R. 1.10 and is addressed to the Assistant Commissioner for Patents, Washington, D.C. 20231.

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METHOD AND APPARATUS FOR CONFERENCE BRIDGE PROCESSING OF SPEECH

Inventors: Eyal Shlomot
Huan-Yu Su
Adil Benyassine
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FIELD OF THE INVENTION

The present invention relates generally to telecommunication systems. In particular, the present invention relates to the processing of speech signals. More particularly, the present invention relates to the processing of speech signals in the context of conference call bridging.

BRIEF DESCRIPTION OF THE DRAWINGS

A more complete understanding of the present invention may be derived by referring to the detailed description when considered in connection with the following Figures:

FIG. 1 is a schematic representation of a conference bridging system in accordance with the present invention;

FIG. 2 is a schematic representation of a conference bridging element in accordance with the present invention;

FIG. 3 is a schematic representation of an exemplary configuration that may be utilized in a practical application; and

FIG. 4 is a flow diagram of an exemplary intelligent bridging process in accordance with the present invention.

DETAILED DESCRIPTION OF PREFERRED EMBODIMENTS OF THE INVENTION

The following description of the preferred exemplary embodiments are not meant to limit the scope of the present invention in any way. Those skilled in the art will recognize that changes and modifications may be made to the preferred embodiments without departing from the scope of the present invention. These and other changes or modifications are intended to be included within the scope of the present invention, as broadly described herein.

Intelligent Mixing of Speech Channels for a Conference Bridge

Description of the problem

A conference bridge enables a conference call, as multiple input speech channels are mixed together and then fed into multiple output speech channels. The input speech channels are digital, and carry the speech information in a coded digital form. The digital form for each channel is a bit stream, generated by a speech encoder at the remote end of the conference call. Each bit stream can be generated by a different speech coding standard, for example, G.711, G.726, G.729(A), or G.723.1 (at two possible bit rates). One possible approach for the mixing of the several speech channels is the decoding of the speech (each channel with its appropriate decoder), summation of the speech signal into a single channel, and re-encoding of the mixed channel (with the appropriate encoders) to generate the bit streams for the multiple output channels.

Several problems are encountered with this direct mixing approach. The problems arise for the case of one active talker at a given time, as well as for the case of multiple active talkers at a given time. First, even for a signal active talker, it is clear that in this approach each speech signal is coded twice, first to generate the bit stream into the conference bridge, and then to generate the bit stream out of the conference bridge. It is well known that this tandem coding result in a degradation of the speech. Another problem arises when several talkers try to talk at the same time. Since low-bit rate speech coders are highly tuned for a single talker (by using, for example, a limited order spectral model and a single pitch representation), they are unsuitable for the coding of a signal that is comprised of several talkers at the same time. Another issue is the computation complexity in the conference bridge. While several speech parameters, such as spectrum, pitch, energy, level of background noise, are known for each individual decoder, they have to be re-computed by the encoder of the mixed signal.

Our solution

We propose the approach of intelligent conference bridge operation. Intelligent bridge comprises of 4 basic steps. At the first step all input speech channels are aligned and a common framing is established, and parameters extraction is performed for channels that use non-parametric coders. The second step involves an intelligent speech mixing of the speech waveform of the input channels, the third step is an intelligent mixing of the parameters of the input speech channels, and the fourth step is an intelligent re-encoding of the mixed output speech channels. These steps can incorporate priority assignment and speech enhancement (for example, by noise reduction or reshaping) for each input and output channel.

This second step and third steps require the modification of the standard speech decoders for their special operation in the conference bridge, and the third and fourth step require the modification of the standard speech encoders for their special operation in the conference bridge.

Framing and Alignment for Speech Mixing in a Conference Bridge

Description of the problem

Several coded speech channels are the input into the conference bridge. The speech at each channel is represented by a bit stream of a speech coding scheme. Not only the format of the bit stream is different from one coding scheme to the other, but also the frame size, for example, from 30 ms in G.723.1 to 20 ms in the futuristic G.4k, to 10 ms in G.729, and to 5 ms for G.728. Moreover, the input bit stream can be coming from a frame-less speech coding approach, such as G.726 or G.711. Intelligent mixing of the speech requires a common frame for the mixing of the parameters.

Our solution

We propose, as a first step in intelligent operation of a conference bridge, the creation of a 'super frame', which is the largest size frame of all of the coding schemes of the speech input channels. For example, if at least one input channel uses the G.723.1 coder, the size of the super frame will be 30 ms. We propose the alignment and the buffering of the short length frames to create a super frame (for example, three 10 ms frames of G.729 to generate a 30 ms super frame suitable for intelligent mixing with G.723.1). We propose the interpolation of the speech parameters from the aligned short length frames to the long length frames, and from the long length frame to the aligned short length frames. We propose creating an aligned super frame structure for the frame-less coding schemes (such as G.711, G.726). We propose parameter extraction and interpolation approach for the non-parametric coders (such as G.711, G.726, and G.728), and the use of these parameters in the intelligent mixing of these coders with other coders.

'Returned-Echo' Cancellation Using Multiple Intelligent Mixing in a Conference Bridge

Description of the problem

A conference call involves several participants. For each participant, the mixed speech information from all the other participants should be provided. One possible solution is the (intelligent) mixing of *all* the channels into a single channel, which is used as the input for each of the output encoders in the conference bridge. The main problem with this approach is that each participant will hear his or her speech, in addition to the speech generated by the other participants. Hearing the speech of oneself, delayed by the two-way digital link and the conference bridge processing time, is perceived as a very annoying returned echo. For an IP based conference bridge, the delay can be of the order of several hundred ms, and the returned echo would be intolerable.

Our solution

We propose the intelligent 'returned-echo' cancellation in a conference bridge. We propose to generate a multiple of mixed signals at the conference bridge, each mixed composed of all the input speech channels, *excluding* the speech of one channel. The mixed signal without the contribution of a particular participant is used as the output speech channel for that particular participant. This mixing scheme removes the contribution of each participant from the signal that is sent back to him/her by the conference bridge, and removes completely the returned echo effect.

Intelligent Spectral Mixing in a Conference Bridge

Description of the problem

The speech spectrum is an important parameter for parametric speech coding. The speech spectrum is commonly represented by the linear prediction (LP) parameters, or by one of their alternative representation, such as normalized autocorrelation function, the reflection coefficients, the arc-sin parameters, the log-area ratios, the line spectral frequencies, the cosines of the line spectral frequencies, as well as the impulse response of the LP filter. Any parametric coder, such as G.723.1 and G.729, transmits a coded representation of the spectrum. It is well known that an accurate representation of the spectrum is crucial for high quality speech, and that the reevaluation of the spectrum is a major source of degradation in tandem coding of speech.

Our solution

We propose to intelligent spectral mixing for the conference bridge. The intelligent spectral mixing uses the decoded spectral information from the mulitple input channels, instead of reevaluating the spectrum of the mixed signal. The spectra can be mixed to provide a meaningful spectral information to the output speech encoder. The spectral mixing can take into consideration the alignment, the framing, the content of each speech input (for example, its energy), as well as timing information, such as a the information about a 'cutting in' talker. The spectral mixing can also be preset to favor specific talker or talkers, providing them a better control over the conference call. The spectral mixing can be performed using any of the representation for the spectrum, described above. In particular, we suggest spectral mixing using the line spectral frequencies (or the cosines of the line spectral frequencies), which are readily available in most parametric coders, in order to reduce the complexity of the conference bridge. We also suggest to obtain a spectral estimate for non-parametric coders, such as G 711, G.726. and G.728, and to use this spectral estimate for the intelligent mixing with parametric coders, such as G.729 and G.723.1.

Intelligent Pitch Mixing in a Conference Bridge

Description of the problem

The pitch is an important parameter in parametric coding of speech. Reevaluating of the pitch from the mixed signal is a simple approach of pitch determination for the mixed output signal in a conference bridge. However, when several participants talk at the same time, the evaluated pitch value might not be meaningful and the mixed signal will be distorted. Moreover, the reevaluation of the pitch will require additional computation for the output channel encoders.

Our solution

We propose an intelligent pitch mixing for the conference bridge. Since each parametric coder, such as G.729 and G.723.1, transmits a description of the pitch we propose to use this pitch information to select a single pitch for the output channel encoders at each time. We propose the mixing of the pitch based on the input channels speech and timing information. We propose this pitch mixing as either a final pitch to be used by the channel output encoders, or as an initial pitch estimate for the channel output encoders. In particular, we propose the selection of a single pitch, based on the energy of the input speech channels and the pitch prediction gain, to be used as an initial estimate for the closed-loop pitch selection, common in low bit-rate coders such as G.723.1 and G.729. We also suggest to obtain a pitch estimate for non-parametric coders, such as G.711, G.726, and G.728, and to use this pitch estimate for the intelligent mixing with parametric coders, such as G.729 and G.723.1.

Priority Assignment in a Conference Bridge

Description of the problem

In a common conference call, the speech signals of all participants are mixed without any priority or preference of one or more participants over the others. Intelligent mixing of speech enables the assignment of a higher or lower priority to one or more participants, which can serve as a tool for managing and controlling the conference call.

Our solution

We propose to use a priority assigning algorithm in intelligent mixing of speech for a conference bridge. A higher or lower priority of a talker can be implemented by a higher or lower weight in mixing his/her speech parameters during parameters mixing, or by a higher or lower level of mixing of the talker speech waveform during the waveform mixing.

Background Noise Handling for a Conference Bridge

Description of the problem

Background noise poses a special problem for low bit-rate speech coders, which are incapable of producing a perceptually faithful representation of most types of background noise. As more and more phone calls are placed from mobile phones, this problem becomes more acute in modern telephony systems. It is well known that the representation of the background noise is worse in tandem coding of speech. In a conference bridge, the representation of the background noise is even more important, since several sources of background noise can be mixed together into a single channel, therefore reducing the overall signal-to-noise ratio.

Our solution

We propose a special approach for background noise handling in a conference bridge. We propose tracking the speech and the background noise activity, the background noise level, and the background noise statistics, for each of the incoming channels. We propose modifying the conference bridge speech decoders and the conference bridge speech encoders to enhance the background noise mixing and representation. We also propose to apply a speech enhancement (for example, by noise reduction) for the input speech channels and/or for the combined mixed waveform, to reduce the particular noise from each channel and the overall noise contribution in the conference bridge.

Conference Bridging Product

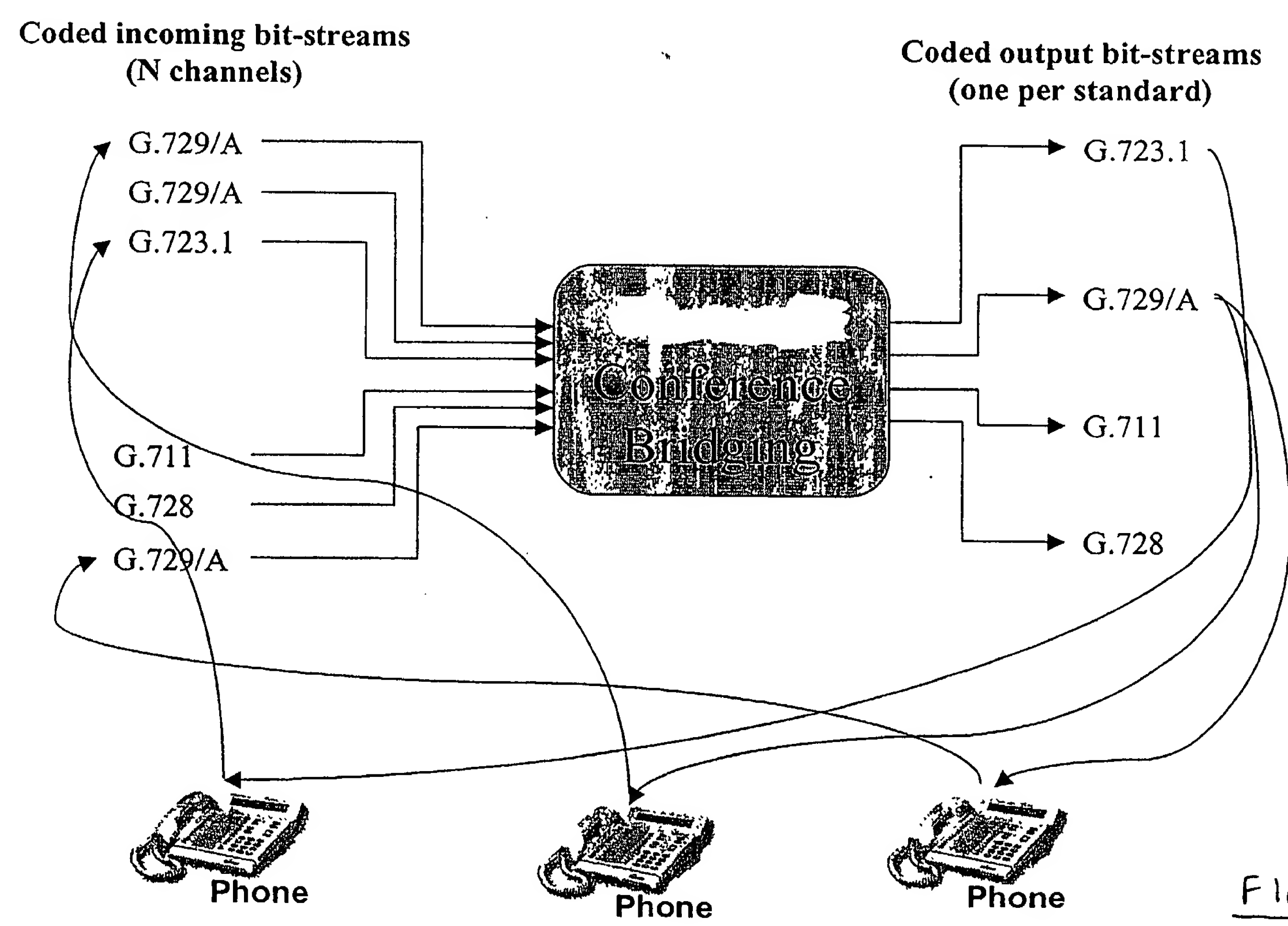


FIG. 1

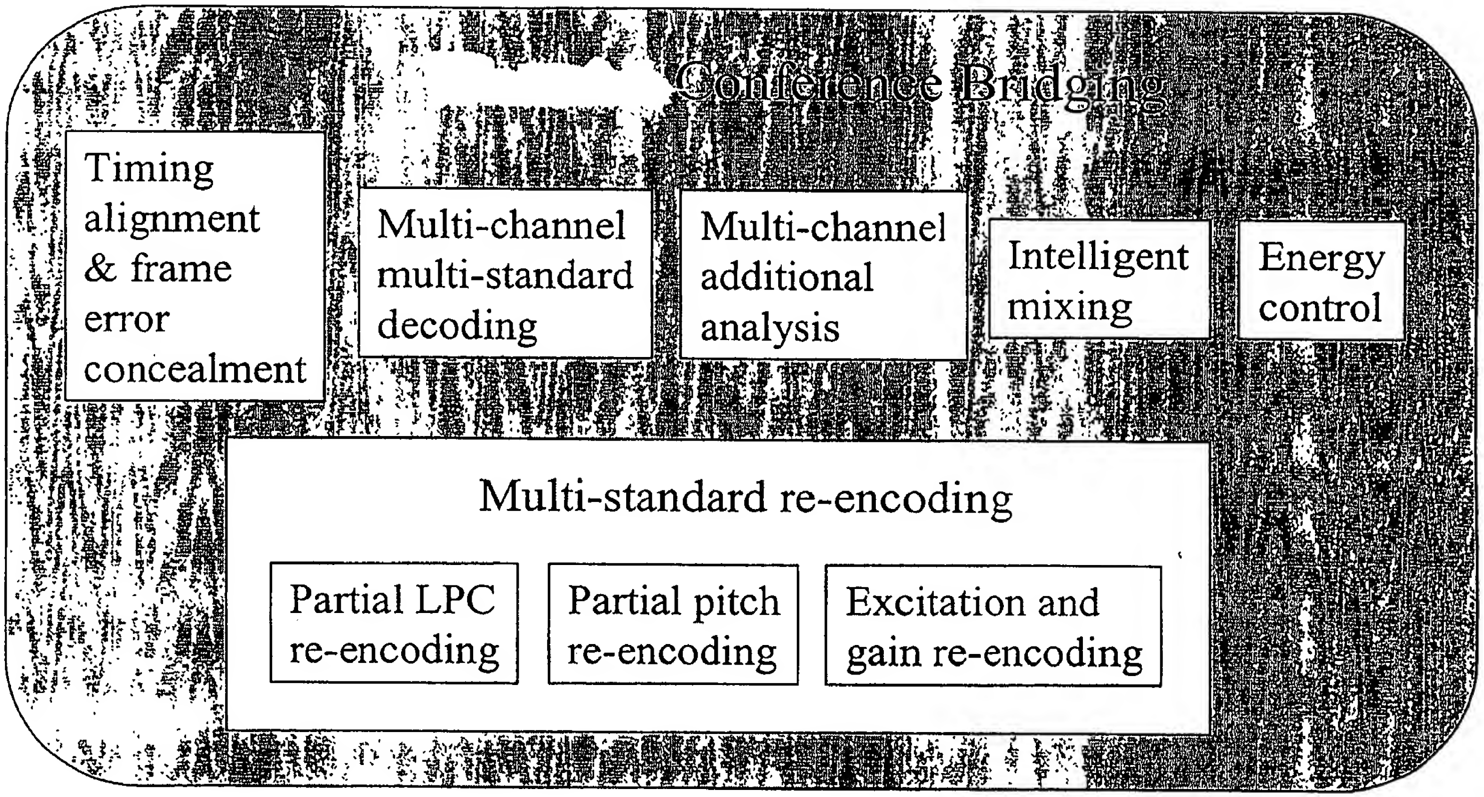
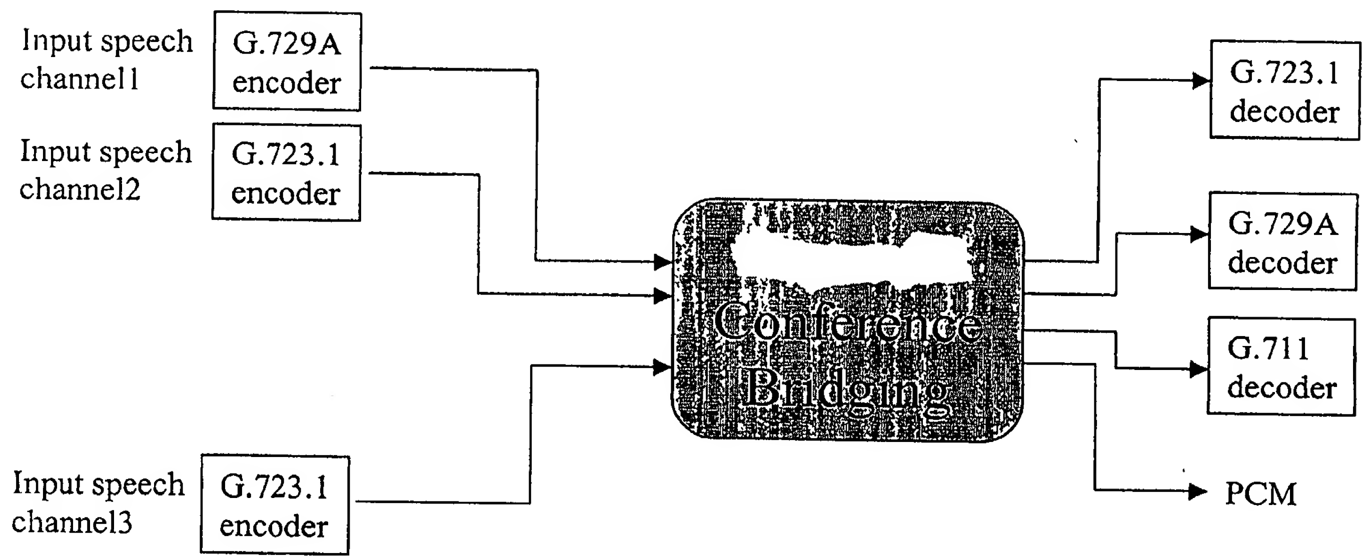

Conference Bridging
Timing alignment & frame error concealment
Multi-channel multi-standard decoding
Multi-channel additional analysis
Intelligent mixing
Energy control
Multi-standard re-encoding
Partial LPC re-encoding
Partial pitch re-encoding
Excitation and gain re-encoding

FIG. 2

Demo configuration



Notes:

1) PCM is the mixed signal without re-encoding which represents the best possibly achievable quality by the conference bridging.
2) G.729A is used instead of G.729
3) The lower rate of G.723.1 (5.3k) is used in the demo.

FIG. 3

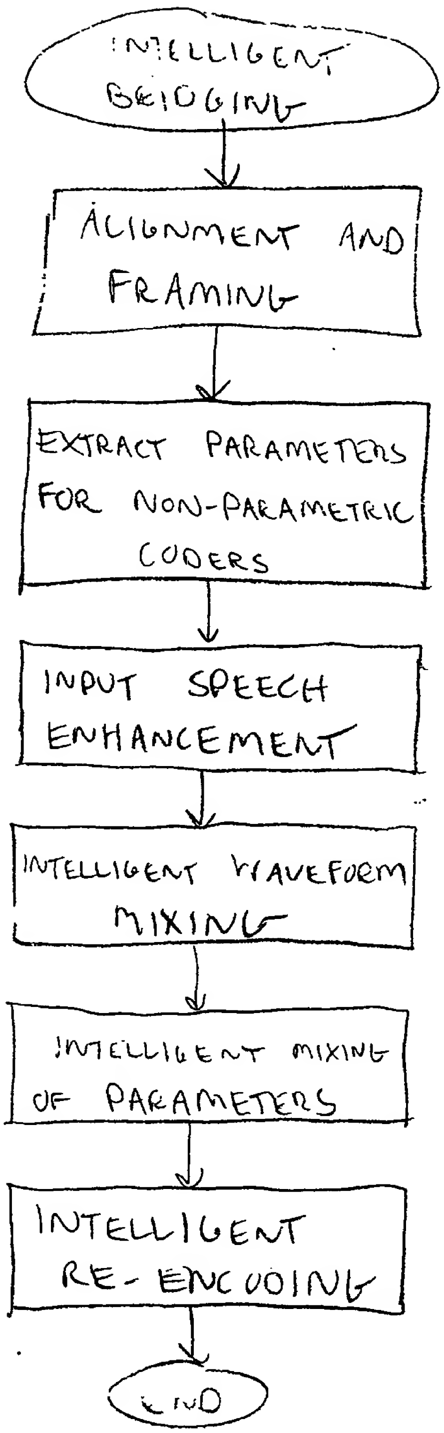

INTELLIGENT BRIDGING
ALIGNMENT AND FRAMING
EXTRACT PARAMETERS FOR NON-PARAMETRIC CODERS
INPUT SPEECH ENHANCEMENT
INTELLIGENT WAVEFORM MIXING
INTELLIGENT MIXING OF PARAMETERS
INTELLIGENT RE-ENCODING
END

FIG. 4